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**REQUEST FOR FILING A
PROVISIONAL PATENT APPLICATION**

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Alexandria, VA 22313-1450

1. This is a request for filing a PROVISIONAL APPLICATION FOR PATENT under 37 C.F.R. 1.53(c).

2. The name and address of the inventor is:

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3. The title of the invention is: TWO-WAY COMMUNICATIONS DEVICE HAVING A SINGLE TRANSDUCER

4. Direct all telephone calls to Stephen S. Ford, Registration No. 35,139, at (503) 222-3613 and send all correspondence to:

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5. The invention was made by an agency of the United States Government or under a contract with an agency of the United States Government.

☒ No

☐ Yes --The name of the U.S. Government agency and the Government contract number are: ____________.

6. Enclosed are the following documents:

☒ Specification (20 pages); Drawings (9 sheets comprising 9 Figures); Claims (14 pages); Abstract (1 page).

7. The filing fee is enclosed:

☒ $160 (large entity) ☐ $80 (small entity)

8. ☒ PTO Form 2038 authorizing credit card payment for the above-listed fees is enclosed.

9. The Commissioner is hereby authorized to charge any additional fees which may be required in connection with the filing of this application, or credit any overpayment, to Account No. 13-1703. A duplicate copy of this sheet is enclosed.

Respectfully Submitted,

By ____________________

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TWO-WAY COMMUNICATIONS DEVICE HAVING A SINGLE TRANSDUCER

BACKGROUND OF THE INVENTION

1. Technical Field of the Invention

This invention relates to a two-way communications device that uses the vibrations of a eardrum membrane caused by a user's voice, and more particularly, to a two-way communications device that has one transducer for receiving and sending voice, and that also serves as a high-level echo canceller (EC) and/or as a high-level voice-operated exchanger (VOX).

2. Description of the Related Art

A generally known device for two-way voice communication is a set consisting of a microphone and an earphone. The microphone and earphone are integrated as a headset so as to make the user's hands free (i.e., the user can use his or her hands freely), but the microphone can also receive noises around the user.

As an alternative, a user's voice is received not at his or her mouth, but through bone conduction, so that the user is made "mouth-free" (i.e., the user can keep the periphery of his or her mouth free). However, a voice received through bone conduction is imperfect and of bad sound quality, although the superposition of peripheral noises can be suppressed to some degree.

Recently, there has been proposed the idea that two transducer elements — one for a microphone and one for an earphone — can be prepared and inserted into a user's right and left ears, respectively, so that the user's voice is detected by vibrations of the membrane of one of the user's eardrums. In this manner, superposition of the noises around the user can be completely suppressed, so that a clear voice having good sound quality can be received by the microphone.

Furthermore, the use of only a single transducer element has been advanced. The idea is to commonly use the single transducer element for both transmission and reception. With this method, only one earplug is needed. Since the user's other ear need not have an earphone, the user may also hear sounds and voices around him or her.

It is inevitable for these devices to have an echo-cancellation (EC) function, namely a function for preventing reception signals from being superimposed on transmission signals. Furthermore, in some cases, there also is a voice-operated exchange function, namely a function for switching between the transmission and reception modes of operation according to the presence or absence of transmission and/or reception signals.

For example, Japanese Patent Application Publication No. 2001-60895 discloses a transmission-and-reception circuit that utilizes an all-analog technique that is equipped with full or partial EC functions and full or partial VOX functions, wherein a bridge circuit, an amplifier, and a comparator are combined with a bridge circuit having an embedded single transducer.

However, with these types of devices it is economically difficult to satisfactorily achieve both the EC and VOX functions. Because this difficulty is particularly significant in the case of the above-mentioned transmission-and-reception circuit having a single transducer element, this device has not yet come into practical use.

When designing an EC function, it is necessary to create a circuit to simulate the actual impedance characteristics of the transducer, namely the characteristics when the transducer is inserted into a particular user's external ear canal, in order to balance the circuit with the actual transducer. However, the transducer has inductive properties. Furthermore, the actual impedance of the transducer varies depending on the individual user, the surrounding environment, and the time.

An ordinary analog-simulation circuit consisting of capacitors (Cs) and resistors (Rs) can be balanced approximately only at a single specific frequency by tuning two variable elements, and therefore it is impossible to achieve balance over the entire frequency range. Although in theory it might be possible to achieve balance over the entire frequency range if an inductance element is included in an analog-simulation circuit, such an inductance element will be bulky and expensive, and it will also be difficult to achieve tuning that conforms to the variations of the characteristics of the transducer, making practical application impossible.

Also, in the case of half-duplex two-way communications, a VOX is essential. The VOX must monitor reception and/or transmission voice(s), then determine in less

than a few milliseconds whether to select the transmission mode or reception mode, based on the continual monitoring of data for the presence or absence of reception and/or transmission signal(s).

The reception and transmission of voice signals not only varies in magnitude continually during a conversation, but at some times it is also intermittently disconnected. Therefore, it is necessary to accumulate and process monitoring data up to the time when a determination is made whether switching should be effected from the reception mode to the transmission mode or vice versa. This determination procedure is difficult, especially when only an analog-circuit technique is employed.

Embodiments of the invention address these and other disadvantages of the conventional art.

SUMMARY OF THE INVENTION

For the purpose of solving the various above-mentioned problems, an embodiment of the invention provides a two-way communications device that has excellent EC and/or VOX functions. Embodiments of the invention are especially for two-way communication devices employing a single transducer element, wherein a microprocessor unit (MPU, and especially a digital circuit such as a digital-signal processor, or DSP), is combined with an analog circuit.

Another embodiment of the invention provides a small and economical two-way communications device having a single transducer that can clearly transmit only a user's voice, without also transmitting noises that are in the user's immediate environment. The embodiment also enables the user to simultaneously hear voices and sounds in his or her immediate environment — other than the voice from the other party being communicated with by the communications device.

Even when reception signals and transmission signals are superimposed in a transducer, embodiments of the invention provide an excellent echo-cancellation function, substantially stopping the reception signal from leaking into the transmission signal (echo) over the entire frequency range.

Furthermore, in the case of half-duplex communications, embodiments of the invention have an excellent VOX function that switches naturally between transmission

and reception modes without entailing unnatural interruption or disconnection of natural conversation.

BRIEF DESCRIPTION OF THE DRAWINGS

Embodiments of the invention will be more readily understood with reference to the following drawings.

FIG. 1 is a block diagram illustrating a full duplex, two-way communications device according to an embodiment of the invention.

FIG. 2 is a block diagram illustrating a full duplex, two-way communications device according to another embodiment of the invention.

FIG. 3 is a block diagram illustrating a half-duplex, two-way communications device according to yet another embodiment of the invention.

FIG. 4 is a block diagram illustrating a half-duplex, two-way communications device according to still another embodiment of the invention.

FIG. 5 is a block diagram illustrating a half-duplex, two-way communications device according to a different embodiment of the invention.

FIG. 6 is a block diagram illustrating a half-duplex, two-way communications device according to another different embodiment of invention.

FIG. 7 is a block diagram illustrating a half-duplex, two-way communications device according to yet another different embodiment of the invention.

FIG. 8 is a block diagram illustrating a half-duplex, two-way communications device according to still another different embodiment of the invention.

FIG. 9(a) is a plot of various gain-transition curves for an attenuator in the VOX of a half-duplex two-way communications device according to embodiments of the invention.

FIG. 9(b) is a table showing the actual results of a sensibility test using the gain-transition curves of FIG. 9(a).

DEFINITIONS

MPU .... Microprocessor unit
EC .... Echo-canceller
VOX .... Voice-operated exchanger

AGC .... Automatic gain controller
L .... Piezoelectric transducer
R1, R2, R3, R4 .... Resistor (also indicating resistance value)
R .... Variable resistor with an intermediate tap "t"
C1, C2 .... Capacitor (also indicating capacitance value)
AMP1, AMP2, AMP3 ... Amplifier
D/A1, DA2, D/A3 .... D/A converter
A/D1, A/D2 .... A/D converter
BUF1, BUF2 .... Buffer
ADD .... Adder
FIL, FIL1, FIL2 .... Filter
k-Calculator, k1-Calculator, k2-Calculator
.... Calculator for filter parameters k, k1, k2
SW1–SW5, SW .... Switch
ATT1, ATT2 .... Attenuator
ATT3, ATT4 .... Analog attenuator
LPF1, FPF2 .... Low-pass filter
Rx .... Reception terminal (also, reception signal)
Tx .... Transmission terminal (also, transmission signal)

DETAILED DESCRIPTION OF THE INVENTION

FIG. 1 is a block diagram illustrating a full duplex, two-way communications device 10 according to an embodiment of the invention. Device 10 includes an analog signal processor (ASP) 105. ASP 105 includes a piezoelectric transducer (or coil L). Device 10 also includes a digital signal processor (DSP) 110. The DSP 110 includes a first digital-to-analog (D/A) converter D/A1, a first analog-to-digital (A/D) converter A/D1, a second A/D converter A/D2, an MPU, and a second D/A converter D/A2. The MPU can be embodied with a single DSP and/or a CPU or a plurality of DSPs and CPUs.

For convenience, and throughout the rest of the detailed description, the analog-to-digital converters found in the various described embodiments may simply be referred to as converter A/D1, converter A/D2, etc, where the number following the "A/D" differentiates between converters found in the same embodiment. Likewise, the digital-to-analog converters found in the various described embodiments may be referred to as converter D/A1, converter D/A2, etc. Using this notation, both the type of converter and the specific converter being described is apparent.

Returning to FIG. 1, the piezoelectric transducer (L) of the ASP 105 is inserted into an external ear canal and functions to convert voltage corresponding to a reception

signal Rx into vibrations (acoustic waves), and to convert the vibrations (acoustic waves) into an electromotive force corresponding to a transmission signal Tx. The piezoelectric transducer is electrically equivalent to an inductive element, and therefore it is represented throughout the disclosure as a coil L in the drawings.

An arrow extending from the coil L toward the external ear canal shows the vibrations (acoustic waves) corresponding to the voltage applied to the coil L, while the other arrow extending in the reverse direction shows air vibrations through the vibrations of an eardrum membrane caused by the user's voice, with the air vibrations generating a corresponding electromotive force within the coil L.

At the reception terminal, the reception signal Rx is processed sequentially through the converter A/D1, the VOX 125, the echo-controller (EC) 120, and the converter D/A1, to be sent to the ASP 105 as reception-signal input. The transmission-signal output of the ASP 105 is processed sequentially through the converter A/D2, the EC 120, the VOX 125, and the converter D/A2, to become a transmission signal at the transmission terminal Tx.

The VOX 125 includes first and second attenuators ATT1 and ATT2, as well as first and second low pass filters LPF1 and LPF2, along with a power controller. The power controller measures the power of the reception signal Rx and the transmission signal Tx in order to control the gains of the first and second attenuators ATT1 and ATT2. This is done to place the respective output of the first and second attenuators ATT1 and ATT2 in a predetermined value range. The VOXs 225, 325, 425, 525, and 625 illustrated in FIGS. 2, 3, 4, 5, and 6, respectively, have the same components as VOX 125 and thus a description of those VOXs will not be repeated.

In alternative embodiments of the invention, the VOX 125 may have the same components as shown in FIG. 1, but may be referred to as an automatic gain controller (AGC).

The Rx and Tx terminals are connected to each other via the piezoelectric transducer L. In normal operation a so-called "echo" is generated, that is, some parts of the reception signal Rx are superimposed on the original transmission signal Tx. It is necessary to suppress such an echo using echo-cancellation techniques. In this

embodiment, the ASP 105 performs a first echo control function, while the echo controller 120 performs a second echo control function.

The ASP 105 is equipped with a bridge circuit 115 that includes the piezoelectric transducer L, resistors R1, R2, R3, and R4, and capacitors C1 and C2.

A first side of the bridge circuit 115 includes the piezoelectric transducer L. The second and fourth sides, which are adjacent to the first side, include the second and fourth resistors R2 and R4, respectively. The connection node between the first and fourth sides is grounded.

The output of the converter D/A1 is sent to the connection node between the second and third sides of bridge circuit 115 as reception-signal input via a first amplifier AMP1, and the differential potential between the connection node between the first and second sides and the connection node between the third and fourth sides is applied to a second amplifier AMP2. The output of AMP2 is transmitted to the converter A/D2 as transmission-signal output of the ASP 105.

Here, the first side of the bridge circuit 115 includes the piezoelectric transducer L in parallel with a series combination of a first resistor R1 and a first capacitor C1. The third side of the bridge circuit 115, which is positioned opposite the first side, includes a third resistor R3 in parallel with a second capacitor C2.

In this circuit configuration, the values of the resistors R1 and R3 are variable and can be controlled so that both the phase and the gain are balanced for a specific value of L (i.e., the value of L when the transducer is inserted into either the right or left external ear canal of a specific user), with respect to at least a single specific frequency.

Thus, superposition of the reception-signal input onto the differential input of the amplifier AMP2 is prevented (echo-cancellation). As a result, only the electromotive force on the transducer appears as the differential input. For example, 600 Hz may be selected as a specific frequency. Using only the bridge circuit 115, however, the leaking of reception signals (i.e., echoes) cannot necessarily be prevented with respect to all the other audible frequencies to either side of the specific frequency.

When the EC 120 is in regular operation, the first and second switches, SW1 and SW2, are both connected to the "r" side of the switch.

During regular operation, a reception signal is output from the EC 120 via a first buffer BUF1, while a transmission signal is output from the converter A/D2 via a second buffer BUF2.

A filter FIL is provided for simulating transmission characteristics from the converter D/A1 to the buffer BUF2 via the ASP 105.

A reception signal is simultaneously transmitted to the converter D/A1 and processed by the filter FIL. The resultant output of the filter FIL is subtracted from the transmission-signal input by an adder ADD, and the resultant output is transmitted to the VOX 125 as the transmission-signal output of the EC 120.

Because the EC 120 can execute the simulation over the entire frequency range, the part of the transmission-signal input that is due to leakage of the reception signal (the echo) is equal to the output of the filter FIL, which means that the adder-function output contains no reception-signal components, such that the echo-cancellation becomes practically perfect.

In the EC 120, the parameter k of the filter FIL (the maximum value of k is equal to the number of taps in the filter, for example, 256) are automatically set by a preinstalled program (not shown). This may be done either immediately after the piezoelectric transducer L is attached to an object such as an external ear canal, and/or periodically when the transducer L is attached in that way, and/or each time that a reception signal and/or a transmission signal is started.

The electrical characteristics of the transducer L, when looked at from the two-way communications device, are slightly different and vary depending on the structure and environment of the particular external ear canal. In other words, the electrical characteristics may vary depending on the temperature and degree of moisture in the external ear canal during communication.

During measurement (test) operation of the EC 120, when the parameters k of the filter FIL are to be set, the switches SW1 and SW2 are connected to the "m" side and the test-signal generator may generate a test signal. The test signal may be a digital signal that corresponds to each impulse, an actual voice produced during conversation, a natural voice, a reception sound, a musical sound, a diffusion code signal, or a tone sweep signal.

Thus, instead of a reception signal from the VOX 125, the generated test signal is transmitted to the converter D/A1 via the buffer BUF1. The resultant output of the converter A/D2 and the test signal are supplied, via the buffers BUF1 and BUF2, respectively, to a k-calculator, so that the values of the parameters k are set in accordance with a predetermined method of calculation.

Also, because the "m" side of the switch SW2 is grounded, the transmission-signal input to the VOX 125 is also grounded, and any output of the adder function during calculation for setting parameters will not leak as noise to the transmission-signal terminal Tx.

If a selected test signal is an impulse, the predetermined calculation process treats the entire frequency range equally, and the calculation is relatively simple. However, if some particular frequency characteristics are required, more desirable echo-cancellation can be achieved by using another test signal and corresponding calculation process. In such calculations, it is necessary to execute complicated calculations such as discrete fast Fourier transforms (FFT) in a shorter time than the period corresponding to that of the maximum frequency of a voice. The invention fully utilizes the progress of technology such that the most current MPU or the like can execute the above-mentioned calculations, even if the MPU is of a super-small size. The power consumption of such an MPU is low enough so that entire embodiments of the invention may be accommodated in an earplug housing.

The VOX 125 receives a reception signal at the first low-pass filter LPF1 that is output from the converter A/D1, and it receives a transmission signal from LPF2 that is output from the EC 120. The VOX 125 then transmits the reception signal to the EC and the transmission signal to the converter D/A2 through the first and second attenuators ATT1 and ATT2, respectively. The VOX 125 measures the power of the reception signal and the transmission signal using a power-controller. The power-controller is also able to control the gains of the attenuators ATT1 and ATT2 so that the powers of the outputs of the attenuators ATT1 and ATT2 match those of the predetermined values.

FIG. 2 is a block diagram illustrating a full duplex, two-way communications device 20 according to another embodiment of the invention. In the following discussion, the portions of FIG. 2 that differ from FIG. 1 will be emphasized, while those

portions that are the same as FIG. 1 are given lesser mention because they function in a similar manner as that explained above for FIG. 1. In other words, the ASP 205 and the EC 220 of FIG. 2 will be emphasized because they differ substantially from the ASP 105 and the EC 120 of FIG. 1.

In the ASP 205, the output of the converter D/A1 is sent to one end of the resistor R1 as reception-signal input via the amplifier AMP1. One end of the piezoelectric transducer L is connected to the other end of the resistor R1, while the other end of piezoelectric transducer L is grounded.

The connection node of the piezoelectric transducer L and the resistor R1 are connected to the positive (+) differential input end of the second amplifier AMP2, while the output of the third converter D/A3 is connected, via a third amplifier AMP3 and a load circuit, to the negative (-) input end of the amplifier AMP2. The output of the amplifier AMP2 is transmitted to the converter D/A2 as the transmission-signal output of the ASP 205.

Furthermore, the load circuit of the ASP 205 includes a serial circuit of resistors R2 and R3, where one end of the resistor R2 is connected to the output of the amplifier AMP3, one end of the resistor R3 is grounded, and the connection node of the resistors R2 and R3 is connected to the negative (-) input end of the amplifier AMP2. Preferably, the resistance values of the resistors are set such that R2 is equal to R1 and R3 is equal to the representative impedance of the piezoelectric transducer L (for example, $R3 = 2\pi * f0 * L$; where $f0 = 600$ Hz and L is the inductance in Henrys).

As shown in FIG. 2, the EC 220 includes first through fifth switches SW1–SW5, all of which are connected at the "r" side of the switch if the EC 220 is in regular operation. The EC 220 further includes first and second filters FIL1 and FIL2.

The EC 220 receives a reception signal that is output from ATT1 of the VOX 225. The reception signal is subsequently buffered by buffer BUF1 and transmitted to the converter D/A1. The EC 220 also transmits the buffered reception signal to the converter D/A3 through the second filter FIL2, and to the adder ADD through the first filter FIL1.

At a second buffer BUF2, the EC 220 receives a transmission signal that is output from the converter A/D2. The output of the first filter FIL1 is subtracted from the

buffered transmission signal at the adder ADD, and the difference is output to the VOX 225 as the transmission-signal output from the EC 220.

The second filter FIL2 is set so that transmission characteristics from the output node of the buffer BUF1, through the filter FIL2, the converter D/A3, the amplifier AMP3, the load circuit (resistors R2 and R3), the amplifier AMP2 (via its negative (-) input end), and the converter A/D2, up to the output node of the buffer BUF2 simulate the transmission characteristics from the output node of the buffer BUF1, through the converter D/A1, the amplifier AMP1, the resistor R1, the amplifier AMP2 (via the one (+) input), and the converter A/D2, up to the output node of the buffer BUF2.

The first filter FIL1 is set in such a manner as to simulate the transmission characteristics from the output node of the first buffer BUF1 up to the second buffer BUF2, via (i) the ASP 205 through two paths (of which one starts at the converter D/A1, while the other path starts at the second filter FIL2 and the converter D/A3), (ii) the differential amplifier AMP2, at which the two paths join, and (iii) the converter A/D2.

In the case of measurement (test) operation of the EC 220, when the parameters of the first and second filters are set, the operation is done in three sequential steps, represented with the switches SW1 to SW5 as first, second, and third steps m1, m2, and m3, respectively. During this operation, the switches are first connected to the m1 terminal, then to the m2 terminal, and lastly to the m3 terminal.

For example, the switch SW1 remains connected to the same terminal for the first, second, and third steps. On the other hand, switch SW4 starts at the m1 terminal for the first step, switches to the m2 terminal for the second terminal, and finally to the m3 terminal for the third step. In any case, during measurement operation of the EC 220, a test signal from a test-signal generator, instead of a reception signal from the VOX 225, is provided to the buffer BUF1. These steps will be described in further detail below.

In the first step, a test signal (not shown) is transmitted to the ASP 205 through the buffer BUF1 and the converter D/A1, whereby the input of the converter D/A3 is grounded through SW4 (a value of zero is input). The resultant output of the buffer BUF2 is stored as Signal 1.

In the second step, the same test signal is transmitted to the ASP 205 through the signal path represented by the buffer BUF1 and the converter D/A3, whereby the input of

the converter D/A1 is grounded through SW3 (a value of zero is input), and the resultant output of the buffer BUF2 is stored as Signal 2.

Signal 1, Signal 2, and the test signal are then processed in a predetermined calculation process by a k2-calculator. This operation sets the parameters k2 of the second filter FIL2.

In the third step, the test signal is transmitted to the ASP 205 through the signal path represented by the buffer BUF1 and the converter D/A1. The test signal is also transmitted to the ASP 205 through the signal path represented by the buffer BUF1, the filter FIL2, and the converter DA/3. The resultant output signal of the buffer BUF2 and the test signal are then processed with another predetermined calculation process by a k1-calculator. This operation sets the parameters k1 of the first filter FIL1.

The second filter FIL2, which is set in the first and the second steps, simulates the case where the input voltage of the amplifier AMP2 has a large amplitude, while the first filter FIL1, which is set in the third step, simulates the case where the input voltage of the amplifier AMP2 has a small amplitude.

In this embodiment, because the EC 220 utilizes a preinstalled program to execute all of the adjustments required for echo-cancellation, there is no need for the ASP 205 to have a bridge circuit, and thus hardware can be simply designed and manufactured without adjustment, and the size of hardware can be easily minimized, all of which will bring large benefits.

Using the processes described above, the EC 220 may automatically set the filters FIL1 and FIL2 immediately after the piezoelectric transducer is inserted into an external ear canal of a user. The EC 220 may also periodically set the filters FIL1 and FIL2 while the transducer is inserted in the ear canal. Alternatively, the filters FIL1 and FIL2 may be set each time that a reception signal or/and a transmission signal is started. In this manner, the variation of acoustic characteristics, including those caused by structural differences between the external ear canals of different users, are reflected in the setting of the filters FIL1 and FIL2.

FIG. 3 is a block diagram illustrating a half-duplex, two-way communications device 30 according to yet another embodiment of the invention. The DSP 310 has a different structure than that of the DSP 110 of FIG. 1, such that the MPU has only a VOX

325 that is necessary for half-duplex communications, and no echo-cancellation function. However, the ASP 305 is equivalent to the ASP 105 of FIG. 1.

In this embodiment, the ASP 305 performs the echo-cancellation function, and the VOX 325 switches between transmission and reception. In this embodiment, it is generally not possible to achieve the same high-quality levels of echo-cancellation such as that produced by the EC 120 of FIG. 1 and EC 220 of FIG. 2, and a slight amount of echo will remain. In this embodiment, however, the VOX 325 reduces gains of the attenuator ATT2 or ATT1 at the time of reception or transmission, respectively, so that for practical purposes the echo may be almost totally suppressed.

As a result, the load to be processed by the MPU is light, which means that more-economical manufacturing is possible. Or, if an equivalent MPU is used, the surplus processing ability can be used for improving the quality of the VOX 325.

The VOX 325 receives and monitors reception signals from the reception terminal Rx via the converter A/D1 and receives and monitors transmission signals via the converter A/D2 (from the previous stage). The VOX 325 determines the presence of a reception signal and/or a transmission signal, and decides whether to switch the operation mode either to reception mode (earphone mode) or to transmission mode (microphone mode). Then, using a predetermined procedure, the VOX 325 processes and then transmits the reception signal to the converter D/A1 (to the next stage), while processing and sending the transmission signal, via the converter D/A2, to the transmission terminal Tx.

As examples of how the VOX 325 determines whether to switch between earphone mode and microphone mode, several scenarios are given below. Each of the examples may be implemented by an installed program (not shown).

As a first example, only the reception signal is monitored. The operation mode is switched to the reception mode if a reception signal is present, and to the transmission mode in the absence of a reception signal.

As a second example, only the transmission signal is monitored. The operation mode is switched to the transmission mode if a transmission signal is present, and to the reception mode in the absence of a transmission signal.

As another example, both the reception and transmission signals may be monitored. The operation mode is switched to the reception mode if only a reception signal is present, and to the transmission mode if only a transmission signal is present.

Alternatively, in the presence of both the reception signal and the transmission signals or the absence of both the reception signal and the transmission signal, the operation mode is set to one mode or the other based on the statistical characteristics of the operation modes explained above.

More particularly, the VOX 325 includes first and second low-pass filters LPF1 and LPF2, first and second attenuators ATT1 and ATT2, and a power-controller. After a reception signal and a transmission signal are processed by the low-pass filters LPF1 and LPF2, either or both of the signals are supplied to the power-controller, and at the same time, both of the signals are supplied to the attenuators ATT1 and ATT2, processed, and then transmitted to the converters D/A1 and D/A2.

At that time, the amplitude value(s) of either or both the reception and transmission signal are averaged (for example, square-averaged, or absolute-value averaged) during a predetermined time period T1 by the power-controller so as to determine the power of each signal, and then the power values are compared with predetermined power threshold(s). In that way, the presence or absence of a reception signal and/or transmission signal is determined, so that the next operation mode can be performed.

If the reception mode is selected, the gain of the attenuator ATT1 is changed towards 1 and the gain of the attenuator ATT2 is changed towards 0. If the transmission mode is selected, the gain of the attenuator ATT1 is changed towards 0 and the gain of the attenuator ATT2 is changed towards 1.

At that time, the cumulative effect of a plurality of decisions made during the time period T1 can be determined. That is to say, when decisions to switch the mode to the reception mode are continued, the gains continues to increase according to a predetermined gain-transition curve, while if the decision is made to switch the mode to the transmission mode, the gains decrease in reverse direction according to the predetermined gain-transition curve.

If the next operation mode is selected on the basis of only one determination of the presence or absence of the signal(s), and if the predetermined time period T1 is short, excess mode switching will frequently occur at every natural momentary pause during conversation. In contrast, if the interval T1 is long, the mode is not successfully switched between transmission and reception. Thus, the solution window (an appropriate range of values) for T1 may not be found, even when the predetermined power thresholds are adjusted as best as possible.

However, according to this embodiment, the gains of the attenuators ATT1 and ATT2 are changed only slightly at every natural momentary pause, and thus switching is actually in effect only after the same decision is made over a number of determinations, resulting in natural and normal switching.

In some embodiments, the shape of the predetermined gain-transition curve can be discrete transitions in the form of an S-shaped staircase. In other words, the gain change per unit decision is small near the final value 0 or 1, while it is large in the intermediate range, giving the staircase an overall S-shape. Examples of gain-transition curves that exhibit this type of S-shape can be seen in S1 and S2 of FIG. 9(a).

FIG. 9 shows the results of an evaluation test of the quality of a communicated voice when various kinds of gain-transition curves are applied to the switching of transmission and reception signals in a certain natural conversation. FIG. 9(a) is a plot of various gain-transition curves for an attenuator in the VOX of a half-duplex two-way communications device according to embodiments of the invention. FIG. 9(b) shows the results of a sensibility evaluation of the quality of a communicated voice using the gain-transition curves of FIG. 9(a), where "A" indicates good, "B" is fair, and "C" is bad.

The power of a voice in natural conversation (here defined as the average of squared amplitudes) is changed as another variable in the evaluation. Also, in this test, the predetermined time period T1 for obtaining an average value of squared amplitudes is set to 10 milliseconds, while the predetermined power threshold is set to 15 dBm0, where the dBm0 units indicate power in dBm (dB referenced to one milliwatt) measured at a zero transmission level point.

According to the evaluation, when linear-staircase-type curves L1 to L4 of FIG. 9(a) are used, the quality of a communicated voice is bad ("C") regardless of the step size

D (delta) of 40 dB to 4 dB. Only when S-shaped staircase type curves S1 and S2 are used can good ("A") or fair ("B") quality can be secured near a normal voice-power level (15 dBmO). However, if the S-shaped staircase type curve S2 is used, where the transition of gain from 0 to 1 exceeds more than 300 m/sec, a slight echo might remain.

FIG. 4 is a block diagram illustrating a half-duplex, two-way communications device 40 according to still another embodiment of the invention. In this embodiment, the EC 420 is equivalent to EC 120 of FIG. 1. The DSP 410 also includes a VOX 425 that is the same as the VOX 325 of FIG. 3. Thus, a half-duplex, two-way communications device 40 of higher echo-cancellation quality than device 30 of FIG. 3 can be obtained.

As possible alternative embodiments, the amplifier AMP2 of the ASPs 105, 305, and 405 of FIGS. 1, 3, and 4, respectively, may be configured by tandemly connecting a differential amplifier having a gain of about 1, an amplifier having a gain of about 600, and a low-pass filter having a low-frequency gain of about 1.

FIG. 5 is a block diagram illustrating a half-duplex, two-way communications device 50 according to a different embodiment of the invention. In this embodiment, the ASP 505 and the DSP 510 are equivalent to ASP 205 and DSP 210 shown in the full-duplex, two-way communications device 20 of FIG. 2. Thus, a half-duplex, two-way communications device 50 of still-higher echo-cancellation quality than device 40 of FIG. 4 may be obtained.

FIG. 6 is a block diagram illustrating a half-duplex, two-way communications device 60 according to another different embodiment of invention. This embodiment provides an ASP 605 that is even more simplified than the ASP 305 of FIG. 3.

In this embodiment, the ASP 605 includes a variable resistor R having an intermediate tap t, whereby the position of the intermediate tap t can be controlled by digital signals. One end of the piezoelectric transducer L is grounded, while the other end is connected to the intermediate tap t.

One end of the variable resistor R receives reception signals, namely the output of a converter D/A1 of the DSP 610 via a first amplifier AMP1. The other end of the variable resistor R is connected to a converter A/D2 of the DSP 610, so as to transmit transmission signals from the ASP 605 via a second amplifier AMP2.

The DSP 610 further includes a VOX 625, a converter A/D1, and a converter D/A2. The converters A/D1 and D/A2 are both equivalent to those shown in FIG. 3, but the power-controller of VOX 625 is further equipped with a third output for controlling the position of the intermediate tap t of the variable resistor R.

That is to say, the DSP 610 provides a VOX 625 that is similar to VOX 325 in FIG. 3, and it further provides an echo-cancellation function together with the position control of the intermediate tap t using the third output of the power-controller. The position of the intermediate tap t is controlled so as to move to the output node of the first amplifier AMP1 in reception mode, and to move to the input node of the second amplifier AMP2 in transmission mode. It is controlled so as to move from one to the other node according to a predetermined tap-position transition curve for switching.

If the position of the intermediate tap t is changed from an existing position to a desired final position (one end or the other end of the variable resistor R), a cumulative effect is provided to plural selections made during the time interval T1, similar to the case of the gain of the attenuators ATT1 and ATT2 that was described above with reference to FIG. 3.

That is to say, if the decision to switch to the reception (or transmission) mode continues, the position continues to change towards the one end (or the other end) according to a predetermined tap-position transition curve, while if the decision is changed to switch to the transmission (or reception) mode, the position changes back towards the other end (or the first end) according to a predetermined tap-position transition curve.

Similar to the gain-transition curves S1 and S2 of FIG. 9(a), the predetermined tap-position transition curves can also be of an S-shaped-staircase type. That is, the tap position change per unit decision is small near the final value 0 or 1, while it is large in the intermediate range. Alternatively, the predetermined tap-position transition curves may include those of the linear-staircase-type, such as curves L1 to L4 of FIG. 9(a).

When translating the gain-transition curves of FIG. 9(a) to a tap position for the tap t of FIG. 6, the attenuator gains (0 dB, -20 dB, -40 dB, etc.) in the vertical axis should be read as the tap-position coordinates of appropriate scales, for example, as one end, the center, and the other end of the variable resistor R.

At the time of selection, in order to switch the reception/transmission mode so that conversation is exchanged naturally, the most appropriate combination among the curves should be adopted for the tap position, for the ATT1 gain, and/or for the ATT2 gain.

FIG. 7 is a block diagram illustrating a half-duplex, two-way communications device 70 according to yet another different embodiment of the invention. This embodiment provides simpler structures to replace both the ASP 305 and the DSP 310 of FIG. 3.

In this embodiment, the ASP 705 includes a variable resistor R having an intermediate tap t. One end of the piezoelectric transducer L is grounded, while the other end is connected to the intermediate tap t. One end of the variable resistor R is directly connected to the reception terminal Rx via a first amplifier AMP1 and a first analog attenuator ATT3. The other end of the variable resistor R is directly connected to the transmission terminal Tx via a second amplifier AMP2 and a second analog attenuator ATT4. The position of the intermediate tap t of the variable resistor R and the gains of the first and the second analog attenuators ATT3 and ATT4 may be controlled by digital signals.

Furthermore, the DSP 710 includes a converter A/D1 and an MPU, and the MPU has a VOX 725 that includes a low-pass filter LPF1 and a power-controller. The VOX 725 receives and monitors reception signals from the reception terminal Rx via the converter A/D1 and the low-pass filter LPF1, determines the presence or absence of a reception signal, decides whether to switch the operation mode either to reception mode (earphone mode) or to transmission mode (microphone mode), and then controls the gains of the first and the second analog attenuators ATT3 and ATT4 as well as the position of the intermediate tap t.

The power-controller averages (for example, square-averages or absolute-value-averages) the amplitude values of the reception signals during a predetermined time period (T1) so as to determine the power of the signals, compares the power values with a predetermined threshold, and thus determines the presence or absence of a reception signal, so that the next operation mode can be selected.

If the reception mode is selected, the gains of the attenuator ATT 3 and the attenuator ATT4 are each changed toward 1 and toward 0, respectively, while the position of the intermediate tap t is changed towards one end of the variable resistor R. If the transmission mode is selected, the above-mentioned gains and position are changed in an opposite manner.

Similar to device 60 of FIG. 6, the transition curves of the gains of the analog attenuators ATT3, ATT4, and the position of the intermediate tap t are selected and decided, respectively, so that communicated voices are most naturally switched.

FIG. 8 is a block diagram illustrating a half-duplex, two-way communications device 80 according to still another different embodiment of the invention. This embodiment provides an even simpler ASP 805 to replace the ASP 705 of FIG. 7.

In this embodiment, the variable resistor R, which has the intermediate tap t in FIG. 7, is replaced by a switch SW. In order not to superimpose on a voice the switching noise of the switch SW if switching from transmission to reception mode, the gain of the second analog attenuator ATT4 is first changed from 1 to 0 according to a predetermined transition curve, then the switch SW is switched from transmission to reception, and finally the gain of the first analog attenuator ATT 3 is changed from 0 to 1 according to a predetermined transition curve.

If switching from reception to transmission mode, it is preferable to perform the switching step outlined above in the reverse order. The position of the switch SW is controlled by a signal from the voice-operated exchanger of the digital signal processor 810.

Still other embodiments of the invention address the problem described below. Generally speaking, the voice picked up at an eardrum, that is, a speaker's voice that is detected via the vibrations of the speaker's eardrum membrane by air transmission, suffers more attenuation at higher frequencies, than does a voice detected at the speaker's mouth. For example, the attenuation can be as high as 10 dB at 2,000 Hz, while there is practically no attenuation at the lower frequencies (up to about 1,000 Hz). Therefore, the voice detected at an eardrum might be of a significantly bad quality, with the result that especially explosive sounds are difficult to hear.

Embodiments of the invention may solve this problem by adding a corrective filter to the DSP. That is, in the processing path for the transmission signal of the VOX 125-625 shown in FIGS. 1-6, the output of the second low-pass filter LPF2 is transmitted, via the above-mentioned corrective filter, to the attenuator ATT2 and to the power control. Frequency characteristics of the gain of the corrective filter are then set so as to balance the above-mentioned difference.

Still other embodiments of the invention solve this attenuation problem by providing a half-duplex, two-way communications device that represent slight modifications of FIGS. 7 and 8. These alternative embodiments also add a corrective filter to the DSPs 710 and 810, along with an additional A/D converter and a D/A converter. In other words, the output of the analog attenuator ATT4 of FIG. 7 and FIG. 8 is transmitted, not directly, but through a converter A/D2 (not shown), through the corrective filter (not shown), through a converter D/A1 (not shown), and finally to the transmission terminal Tx.

Embodiments of the invention will now be described in a non-limiting way.

A two-way communications device according to embodiments of the invention are small and economical, but nevertheless provide numerous advantageous features. A voice can be transmitted and received clearly, even in an environment that is very noisy and/or under adverse conditions such as strong wind and rain. Because this device uses only one of a user's ears, the user can freely use his or her hands, mouth, and the other ear. Therefore, even when a person is using this device, he or she can hear and talk with people who are nearby, and can hear sounds emitted from machines around him or her. Accordingly, this device is the most suitable for complicated and dangerous work, for example, that associated with operation of a vehicle or machine.

In light of the above descriptions of the preferred embodiments of this invention, it will be apparent to one of skill in the art that other embodiments incorporating the above described concepts may also be created. Thus, the embodiments of the invention are not limited to the embodiments disclosed above, but rather should be limited only by the spirit and scope of the invention as defined in the following claims.

CLAIMS

1. A two-way communications device comprising:

an analog signal processor that includes a piezoelectric transducer; and

a digital signal processor that includes a first A/D converter, a reception terminal, a transmission terminal, and a microprocessor unit that has a voice-operated exchanger.

2. The device of claim 1, wherein the digital signal processor further includes:

a first D/A converter;

a second D/A converter; and

a second A/D converter, wherein the device is configured to sequentially process a signal at the reception terminal by the first A/D converter, the voice-operated exchanger, the first D/A converter, the analog signal processor, the second A/D converter, and again by the voice-operated exchanger before the signal arrives at the transmission terminal.

3. The device of claim 2, wherein the microprocessor unit further comprises:

an echo canceller coupled to the voice-operated exchanger, the first D/A converter, and the second A/D converter.

4. The device of claim 3, wherein the echo canceller comprises:

a first buffer with an output coupled to both an input of the first D/A converter and an input of a first filter;

a second buffer with an input coupled to an output of the second A/D converter, wherein the filter is configured to simulate transmission characteristics along a signal path starting at the input of the first D/A converter, passing through the analog signal processor, passing through the second A/D converter, and ending at an output of the second buffer;

an adder configured to subtract an output of the filter from the output of the second buffer;

a test signal generator;

a first switch configured to selectively connect an input of the first buffer to either the voice-operated exchanger or the test signal generator;

a parameter calculator coupled to the outputs of the first and second buffers and configured to set the parameters of the first filter by processing a signal from the second buffer and a test signal from the first buffer; and

a second switch configured to selectively connect an output of the adder to either the voice-operated exchanger or to a ground.

5. The device of claim 4, wherein the analog signal processor further comprises:

a four-sided bridge circuit with a node between each of the sides, wherein a first side includes the piezoelectric transducer, and wherein the node between the first and the fourth side is grounded;

a first amplifier coupled between the first D/A converter and the node between the second and third sides; and

a second amplifier coupled to the second A/D converter, wherein the node between the first and the second sides is connected to a differential input of the second amplifier, and wherein the node between the third and fourth sides is connected to another differential input of the second amplifier.

6. The device of claim 5, wherein the bridge circuit further comprises:

on the first side, a series circuit that includes a first resistor and a first capacitor, wherein the series circuit is connected in parallel with the piezoelectric transducer, and wherein the first resistor is selected from the group consisting of a fixed value resistor and a variable resistor;

on the second side, a second resistor;

on the third side, a third resistor connected in parallel with a second capacitor, wherein the third resistor is selected from the group consisting of a fixed value resistor and a variable resistor; and

on the fourth side, a fourth resistor.

7. The device of claim 3, wherein the digital signal processor additionally comprises a third D/A converter coupled between the analog signal processor and the echo canceller.

8. The device of claim 7, wherein the echo canceller comprises:

a test signal generator;

a first buffer;

a first switch configured to selectively connect an input of the first buffer to either the voice-operated exchanger or the test signal generator;

a first filter with an input connected to an output of the first buffer;

a first parameter calculator with a first input coupled to the output of the first buffer and configured to set the parameters of the first calculator;

a second filter with an input connected to the output of the first buffer;

a second parameter calculator with a first input coupled to the output of the first buffer and configured to set the parameters of the second filter;

a second switch configured to selectively connect the output of the first buffer to either an input of the first D/A converter or to ground;

a second buffer with an input coupled to an output of the second A/D converter;

an adder configured to subtract an output of the first filter from an output of the second buffer;

a third switch configured to selectively connect an output of the adder to either the voice-operated exchanger or to ground;

a fourth switch configured to selectively connect an input of the third D/A converter to either the output of the first buffer, the output of the first filter, or to ground; and

a fifth switch configured to selectively connect the output of the second A/D converter to either a second input of the first parameter calculator, a second input of the second parameter calculator, or to an input of the adder.

9. The device of claim 8, wherein the analog signal processor further comprises:

a first amplifier with an input coupled to an output of the first D/A converter;

a second amplifier with an output coupled to an input of the second A/D converter;

a third amplifier with an input coupled to an output of the third D/A converter;

a resistor that is coupled between an output of the first amplifier and an end of the no more than one inductor, wherein another end of the no more than one inductor is grounded, and wherein a node between the resistor and the inductor is coupled to a positive differential input of the second amplifier, and

a load circuit composed of two resistors, wherein a node between the resistors is coupled to a negative differential input of the second amplifier.

10. The device of claim 2, wherein the analog signal processor further comprises:

a four-sided bridge circuit with a node between each of the sides, wherein a first side includes the piezoelectric transducer, and wherein the node between the first and the fourth side is grounded;

a first amplifier coupled between the first D/A converter and the node between the second and third sides; and

a second amplifier coupled to the second A/D converter, wherein the node between the first and the second sides is connected to a differential input of the second amplifier, and wherein the node between the third and fourth sides is connected to another differential input of the second amplifier.

11. The device of claim 10, wherein the bridge circuit further comprises:

on the first side, a series circuit that includes a first resistor and a first capacitor, wherein the series circuit is connected in parallel with the piezoelectric transducer, and wherein the first resistor is selected from the group consisting of a fixed value resistor and a variable resistor;

on the second side, a second resistor;

on the third side, a third resistor connected in parallel with a second capacitor, wherein the third resistor is selected from the group consisting of a fixed value resistor and a variable resistor; and

on the fourth side, a fourth resistor.

12. The device of claim 2, wherein the analog signal processor further comprises:

a first amplifier with an input coupled to an output of the first D/A converter;

a second amplifier with an output coupled to an input of the second A/D converter;

a variable resistor that is coupled between an output of the first amplifier and an output of the second amplifier, wherein the variable resistor has an intermediate tap that is coupled to an end of the no more than one inductor, and wherein another end of the no more than one inductor is grounded.

13. The device of claim 12, wherein the intermediate tap is configured so that a position of the intermediate tap is controlled by a digital signal received from the voice-operated exchanger.

14. The device of claim 1, wherein the analog signal processor comprises:

a first analog attenuator with an input coupled directly to the reception terminal, wherein the first analog attenuator is configured so that its gain is controlled by a digital signal received from the voice-operated exchanger;

a second analog attenuator with an output coupled directly to the transmission terminal, wherein the second analog attenuator is configured so that its gain is controlled by a digital signal received from the voice operated exchanger;

a first amplifier with an input coupled to an output of the first analog attenuator; and

a second amplifier with an output coupled to an input of the second analog attenuator.

15. The device of claim 14, wherein the analog signal processor further comprises:

a variable resistor that is coupled between an output of the first amplifier and an output of the second amplifier, wherein the variable resistor has an intermediate tap that is coupled to an end of the no more than one inductor, and wherein another end of the no more than one inductor is grounded.

16. The device of claim 15, wherein the intermediate tap is configured so that a position of the intermediate tap is controlled by a digital signal received from the voice-operated exchanger.

17. The device of claim 14, wherein the analog signal processor further comprises:

a switch configured to couple an end of the no more than one inductor either to an output of the first amplifier or an input of the second amplifier, wherein another end of the no more than one amplifier is grounded.

18. The device of claim 17, wherein the switch is configured so that a position of the switch is controlled by a signal received from the voice operated exchanger.

19. In a two-way communication device comprising a digital signal processor and an analog signal processor with a transducer that is designed to be inserted into an ear canal, a method comprising:

configuring a filter in the digital signal processor to simulate a signal path through the analog signal processor; and

subtracting an output of the filter from an output of the analog signal processor to substantially cancel an echo component present in the output of the analog signal processor.

20. The method of claim 19, wherein configuring the filter in the digital signal processor to simulate the signal path through the analog signal processor comprises:

generating a test signal;

propagating the test signal through the signal path while the transducer is placed in an external ear canal; and

setting parameters of the filter based on characteristics of the propagated test signal.

21. The method of claim 20, wherein generating the test signal comprises:

generating a test signal chosen from the group consisting of a digital signal that corresponds to any one of the following: an impulse, an actual voice during conversation, a natural voice, a reception sound, or a musical sound; a diffusion code signal, and a tone sweep signal.

22. The method of claim 19, further comprising:

reconfiguring the filter after a predetermined amount of time so that a variation of the acoustic conditions of the ear canal are adjusted for.

23. In a two-way communication device comprising a digital signal processor and an analog signal processor, the analog signal processor including a transducer that is designed to be inserted into an ear canal, a method comprising:

simulating a combined first and second signal path through the analog signal processor using a first filter located in the digital signal processor;

simulating the first signal path through the analog signal processor using a second filter located in the digital signal processor; and

subtracting an output of the first filter from an output of the analog signal processor to substantially cancel an echo component present in the output of the analog signal processor.

24. The method of claim 23, wherein simulating the combined first and second signal path through the analog signal processor using the first filter located in the digital signal processor comprises:

placing the single transducer in an external ear canal;

generating a test signal;

propagating the test signal through the first and second signal paths; and

setting parameters of the first filter based on characteristics of the propagated test signal.

25. The method of claim 24, wherein simulating the first signal path through the analog signal processor using the second filter located in the digital signal processor comprises:

propagating the test signal through the first signal path while an input to the second signal path is grounded;

propagating the test signal through the second signal path while an input to the first signal path is grounded; and

setting parameters of the second filter based on characteristics of the propagated test signal.

26. The method of claim 24, wherein generating the test signal comprises:

generating a test signal chosen from the group consisting of a digital signal that corresponds to any one of the following: an impulse, an actual voice during conversation, a natural voice, a reception sound, or a musical sound; a diffusion code signal, and a tone sweep signal.

27. The method of claim 23, further comprising:

reconfiguring the first filter and the second filter after a predetermined amount of time so that a variation of the acoustic conditions of the ear canal are adjusted for.

28. In a two-way communication device comprising a digital signal processor and an analog signal processor, the analog signal processor including a transducer that is designed to be inserted into an ear canal, the digital signal processor having a voice operated exchanger, a method comprising:

selectively switching between a reception mode and a transmission mode in response to a natural ebb and flow of conversation.

29. The method of claim 28, wherein selectively switching between the reception mode and the transmission mode comprises:

monitoring a reception signal from an input of the two-way communication device;

operating in the reception mode if the reception signal is determined to be present; and

operating in the transmission mode if the reception signal is determined to be absent.

30. The method of claim 28, wherein selectively switching between the reception mode and the transmission mode comprises:

monitoring a transmission signal from an output of the transducer;

operating in the transmission mode if the transmission signal is determined to be present; and

operating in the reception mode if the transmission signal is determined to be absent.

31. The method of claim 28, wherein selectively switching between the reception mode and the transmission mode comprises:

monitoring a reception signal from an input of the two-way communication device;

monitoring a transmission signal from an output of the transducer;

operating in the reception mode if only the reception signal is determined to be present;

operating in the transmission mode if only the transmission signal is determined to be present;

statistically selecting either the reception mode or the transmission mode if both the reception and the transmission signal or neither the reception signal nor the transmission signal are determined to be present.

32. The method of claim 28, wherein selectively switching between the reception mode and the transmission mode comprises:

calculating an average amplitude value over a predetermined time period from at least one signal chosen from the group consisting of a reception signal from an input of the two-terminal device and a transmission signal from an output of the transducer;

determining the presence or absence of the at least one signal by comparing a power level calculated with the average amplitude value to a predetermined threshold;

switching from transmission mode to reception mode by changing a gain of a first attenuator associated with the reception signal from a lower limit to an upper limit and changing a gain of a second attenuator associated with the transmission signal from the upper limit to the lower limit; and

switching from reception mode to transmission mode by changing the gain of the first attenuator from the upper limit to the lower limit and changing the gain of the second attenuator from the lower limit to the upper limit.

33. The method of claim 32, wherein switching from transmission mode to reception mode comprises:

gradually increasing the gain of the first attenuator from the lower limit towards the upper limit according to a predetermined gain transition curve, wherein the gain of the first attenuator becomes closer to the upper limit for every predetermined time interval that the reception mode is indicated; and

gradually decreasing the gain of the second attenuator from the upper limit towards the lower limit according to the predetermined gain transition curve, wherein the gain of the second attenuator becomes closer to the lower limit for every predetermined time interval that the reception mode is indicated.

34. The method of claim 33, wherein switching from reception mode to transmission mode comprises:

gradually decreasing the gain of the first attenuator from the upper limit towards the lower limit according to the predetermined gain transition curve, wherein the gain of

the first attenuator becomes closer to the lower limit for every predetermined time interval that the transmission mode is indicated; and

gradually increasing the gain of the second attenuator from the lower limit towards the upper limit according to the predetermined gain transition curve, wherein the gain of the second attenuator becomes closer to the upper limit for every predetermined time interval that the transmission mode is indicated.

35. The method of claim 34, wherein the predetermined gain transition curve is has a substantially S-shaped staircase profile, wherein the gain change per unit decision is small near the upper and lower limit and large in an intermediate range between the upper and lower limits.

36. The method of claim 35, wherein the upper and lower limits are 1 and 0, respectively.

37. A two-way communications device comprising:

an analog signal processor that includes a piezoelectric transducer and a variable resistor having an intermediate tap; and

a digital signal processor that includes a microprocessor unit and an A/D converter,

wherein the piezoelectric transducer is configured to convert vibrations into an electromotive force and to convert voltage into vibrations,

wherein a first end of the piezoelectric transducer is connected to the intermediate tap,

wherein a second end of the piezoelectric transducer is grounded,

wherein a first end of the variable resistor is connected to a reception terminal via a first amplifier and a first analog attenuator, while a second end of the variable resistor is connected to a transmission terminal via a second amplifier and a second analog attenuator,

wherein a position of the intermediate tap and a gain of the first and the second analog attenuators are controlled by digital signals,

wherein the microprocessor unit has a voice-operated exchanger,

wherein the voice-operated exchanger monitors a reception signal via the A/D converter, determines the presence or absence of reception signals so as to decide the next potential operation mode as either reception mode or transmission mode, and correspondingly controls the position of the intermediate tap and the gains of the first and second analog attenuators.

38. The device of claim 37, wherein:

the voice-operated exchanger comprises a first low-pass filter and a power-controller;

the reception signal is supplied to the power-controller after being processed by the first low-pass filter;

the power-controller is configured to average an amplitude of the reception signal over a predetermined time to determine a power of the reception signal, to compare the power with a predetermined threshold, and to determine the presence or absence of reception signals;

if the mode selected is the reception mode, the device is configured so that a gain of the first and the second analog attenuators are changed towards 1 and 0, respectively, while the position of the intermediate tap is simultaneously changed towards the first end of the variable resistor; and

if the mode selected is the transmission mode, the device is configured so that the gain of the first and second attenuators are changed towards 0 and 1, respectively, while the position of the intermediate tap is simultaneously changed toward the second end of the variable resistor.

39. The device of claim 38, wherein the gain of the first and second attenuators follow a predetermined gain transition curve, wherein the position of the intermediate tap follows a predetermined tap position curve, and wherein the gains and the intermediate tap move incrementally once every predetermined time period.

40. The device of claim 39, wherein the predetermined gain transition curve and the predetermined tap position transition curve both have substantially S-shaped staircase profiles with small gain changes per unit and small tap position changes per unit near the endpoints but large gain changes per unit and large tap position changes per unit in an intermediate range.

41. The device of claim 40, wherein the voice-operated exchanger controls the gain of the first and second analog attenuators.

42. The device of claim 38, further comprising:

a correction filter interposed between a second low-pass filter and the second analog attenuator, the correction filter configured to balance the difference in frequency characteristics between the case when the user's voice is detected via the user's eardrum vibrations, and the case when the voice is detected via the user's mouth.

43. The device of claim 42, further comprising:

a second A/D converter interposed between the second analog attenuator and the correction filter; and

a first D/A converter interposed the correction filter and the transmission terminal.

44. A two-way communications device comprising:

a piezoelectric transducer, wherein the piezoelectric transducer is configured to detect vibrations of an eardrum membrane caused by sound waves, and wherein the piezoelectric transducer is also configured to transmit a sound wave to the eardrum membrane;

a housing shaped like an earplug that is configured to contain the piezoelectric transducer; and

an echo-canceller configured to model the variable acoustic characteristics of the eardrum membrane and an ear canal associated with the eardrum membrane.

45. In a two-way communications device comprising a microphone and a earphone, a method comprising:

generating ultrasonic waves of a predetermined constant frequency using a first piezoelectric transducer;

directing the ultrasonic waves towards an eardrum membrane;

receiving reflected ultrasonic waves from the eardrum membrane with the first or a second piezoelectric transducer;

analyzing a Doppler-effect modulation of the reflected ultrasonic waves caused by vibration of the eardrum membrane;

demodulating the reflected ultrasonic waves to obtain a voice-transmission signal;

generating a sound wave corresponding to a voice-reception signal; and

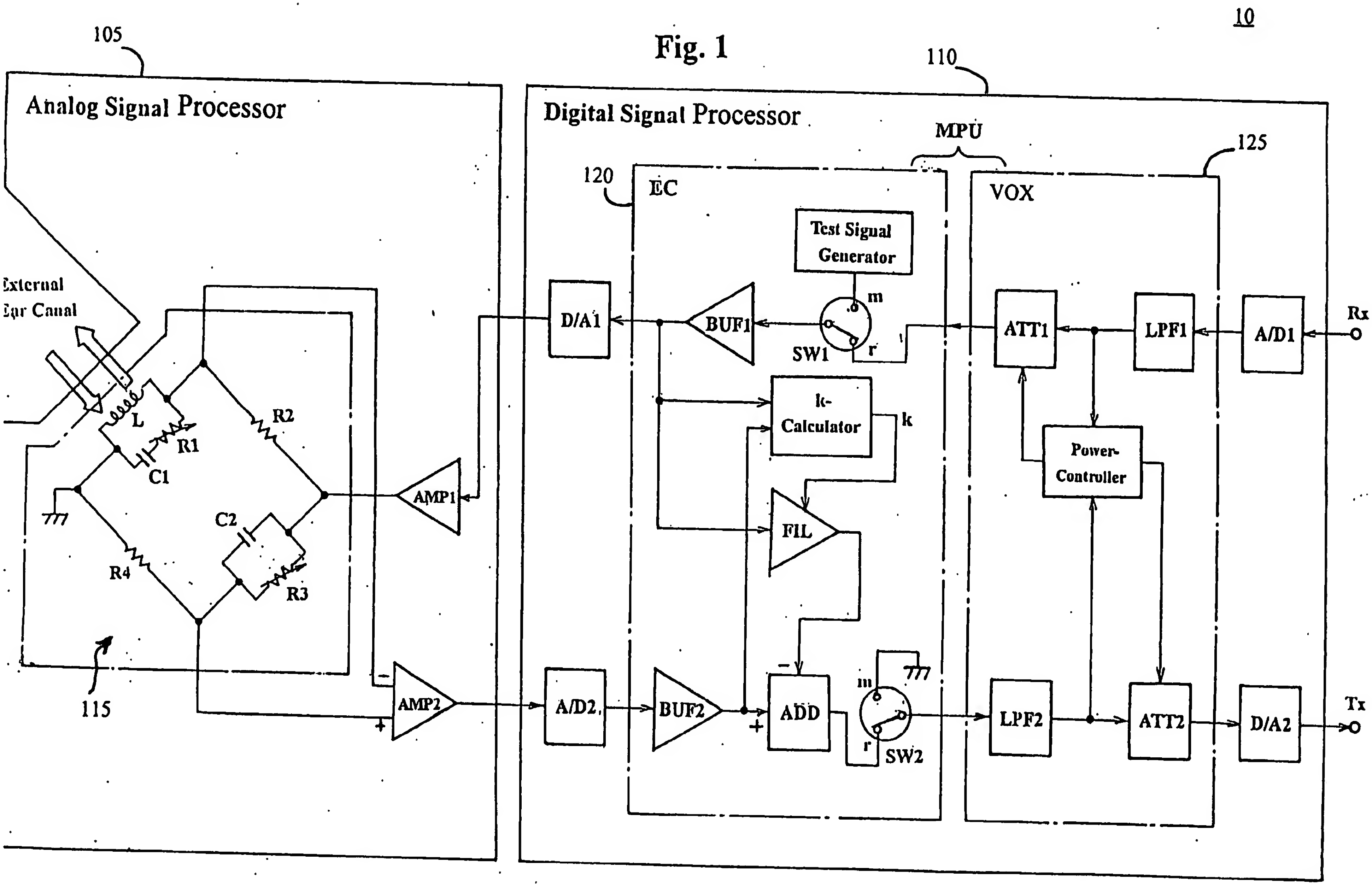
superimposing the sound wave on the ultrasonic waves.

TWO-WAY COMMUNICATIONS DEVICE HAVING A SINGLE TRANSDUCER

ABSTRACT OF THE DISCLOSURE

Embodiments of the invention provide a small and economical two-way communications device that has both an excellent echo-cancellation function that substantially suppresses echoes over the entire frequency range and an excellent voice-operated exchange function that provides natural switching of conversation sounds while protecting against unnatural disconnection or echoes during conversation, even when a reception signal and a transmission signal are superimposed in a single transducer.

Fig. 1



10
105
Analog Signal Processor
External
Ear Canal
L
R1
C1
R2
C2
R3
R4
115
AMP1
AMP2
110
Digital Signal Processor
MPU
120
EC
125
VOX
Test Signal
Generator
m
r
SW1
BUF1
D/A1
k-
Calculator
k
FIL
A/D2
BUF2
ADD
SW2
ATT1
LPF1
A/D1
Rx
Power-
Controller
LPF2
ATT2
D/A2
Tx

20

Fig. 2

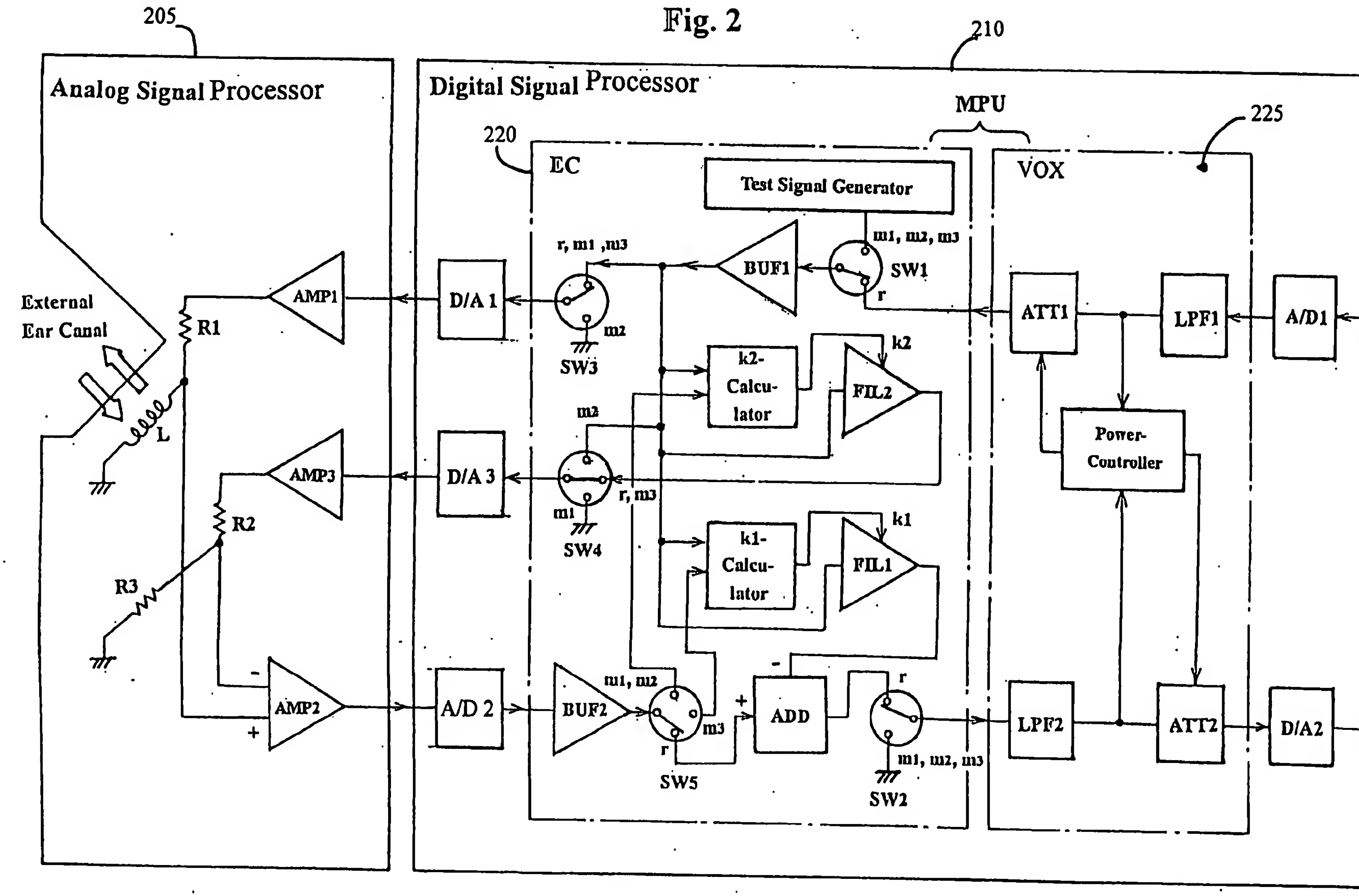
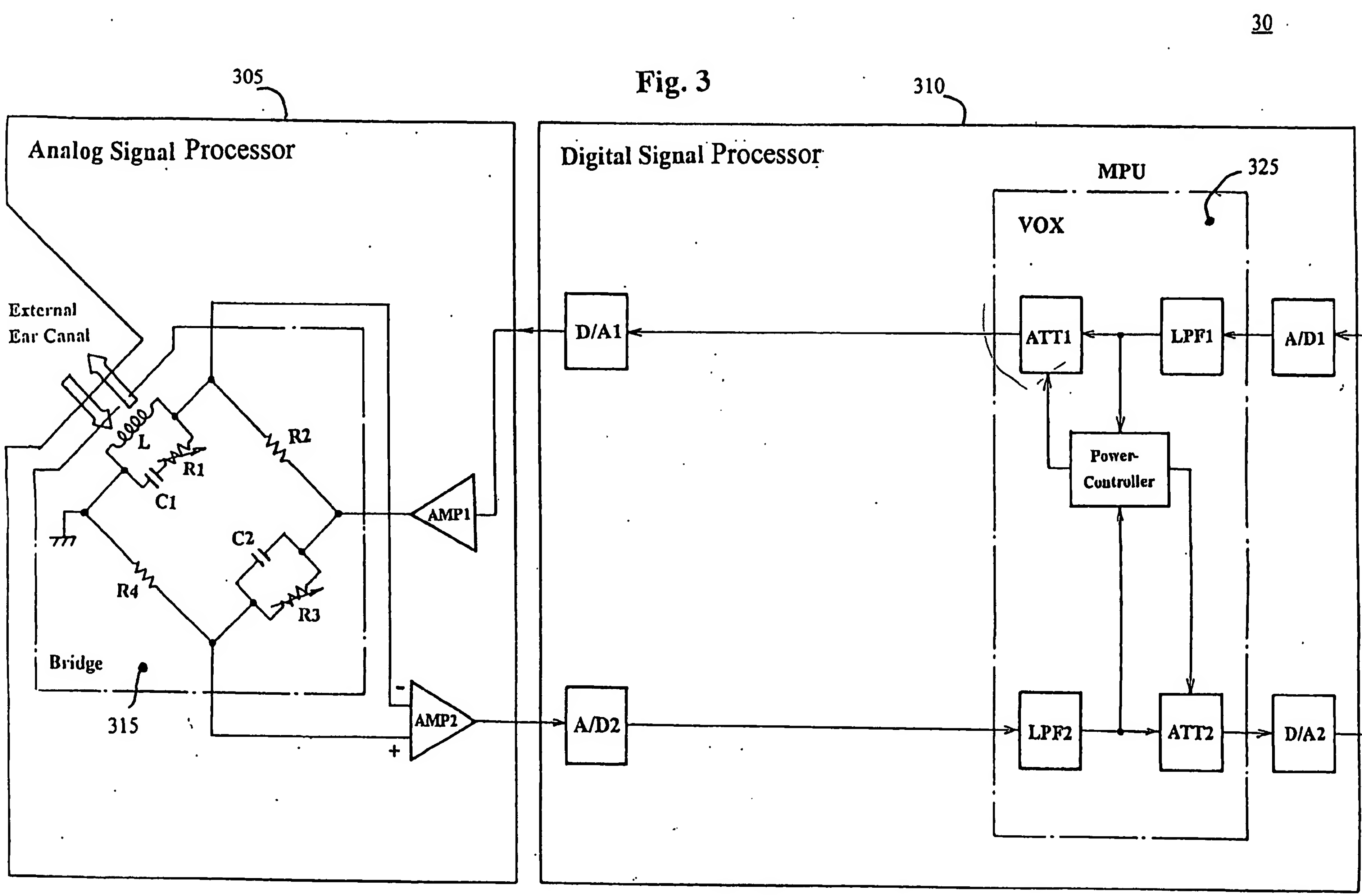
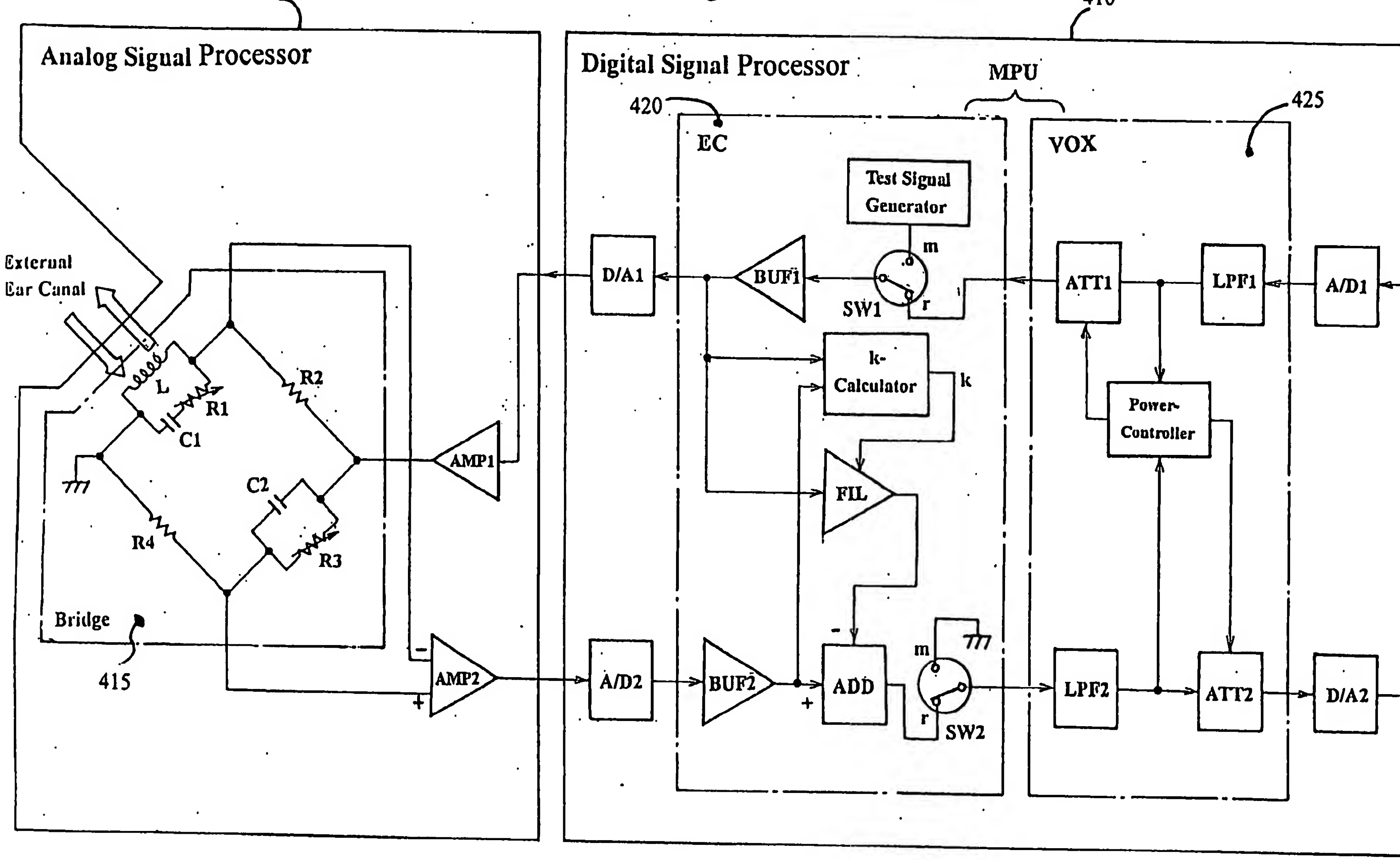

205
210
220
225
Analog Signal Processor
Digital Signal Processor
MPU
EC
VOX
Test Signal Generator
External Ear Canal
AMP1
AMP2
AMP3
R1
R2
R3
L
D/A 1
D/A 3
A/D 2
BUF1
BUF2
SW1
SW2
SW3
SW4
SW5
r, m1 ,m3
m1, m2, m3
r
m2
m1
r, m3
m1, m2
m3
k2-Calculator
k1-Calculator
FIL2
FIL1
k2
k1
ADD
ATT1
ATT2
LPF1
LPF2
A/D1
D/A2
Power-Controller
Rx
Tx

Fig. 3


30
305
310
Analog Signal Processor
Digital Signal Processor
MPU
325
VOX
External
Ear Canal
L
R1
C1
R2
C2
R3
R4
Bridge
315
AMP1
AMP2
-
+
D/A1
A/D2
ATT1
LPF1
A/D1
Rx
Power-
Controller
LPF2
ATT2
D/A2
Tx

40

Fig. 4


405
Analog Signal Processor
External
Ear Canal
L
R1
C1
R2
C2
R3
R4
Bridge
415
AMP1
AMP2
410
Digital Signal Processor
MPU
420
EC
425
VOX
Test Signal
Generator
m
r
SW1
BUF1
D/A1
k-
Calculator
k
FIL
ADD
BUF2
A/D2
SW2
ATT1
LPF1
A/D1
Rx
Power-
Controller
LPF2
ATT2
D/A2
Tx

50

Fig. 5

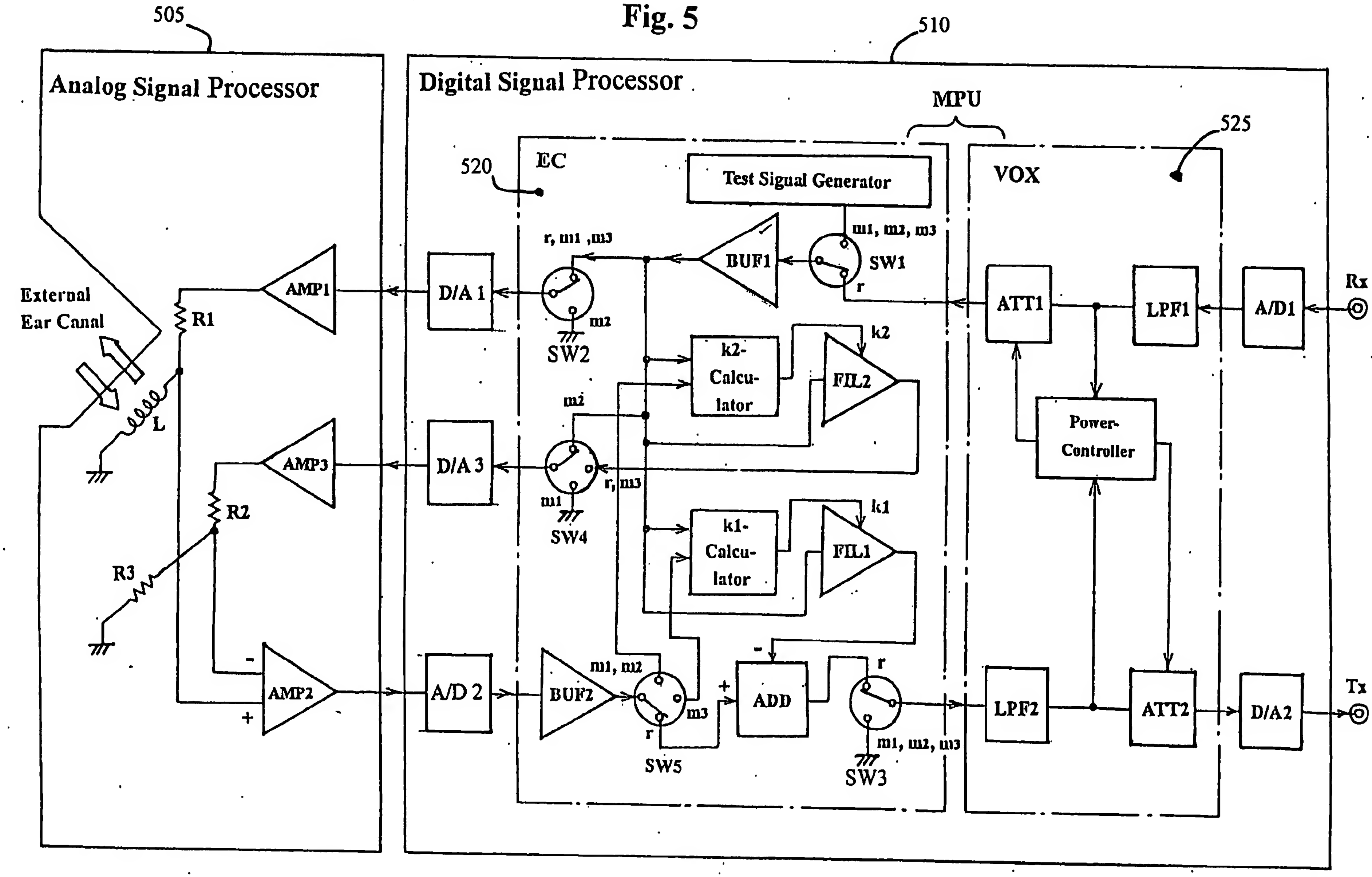

505
510
Analog Signal Processor
Digital Signal Processor
MPU
525
520
EC
VOX
Test Signal Generator
External
Ear Canal
R1
R2
R3
L
AMP1
AMP3
AMP2
D/A 1
D/A 3
A/D 2
r, m1 ,m3
m2
SW2
m2
m1
r, m3
SW4
BUF1
SW1
m1, m2, m3
r
k2-
Calcu-
lator
FIL2
k2
k1-
Calcu-
lator
FIL1
k1
BUF2
m1, m2
m3
r
SW5
ADD
SW3
m1, m2, m3
ATT1
LPF1
A/D1
Rx
Power-
Controller
LPF2
ATT2
D/A2
Tx

Fig. 6

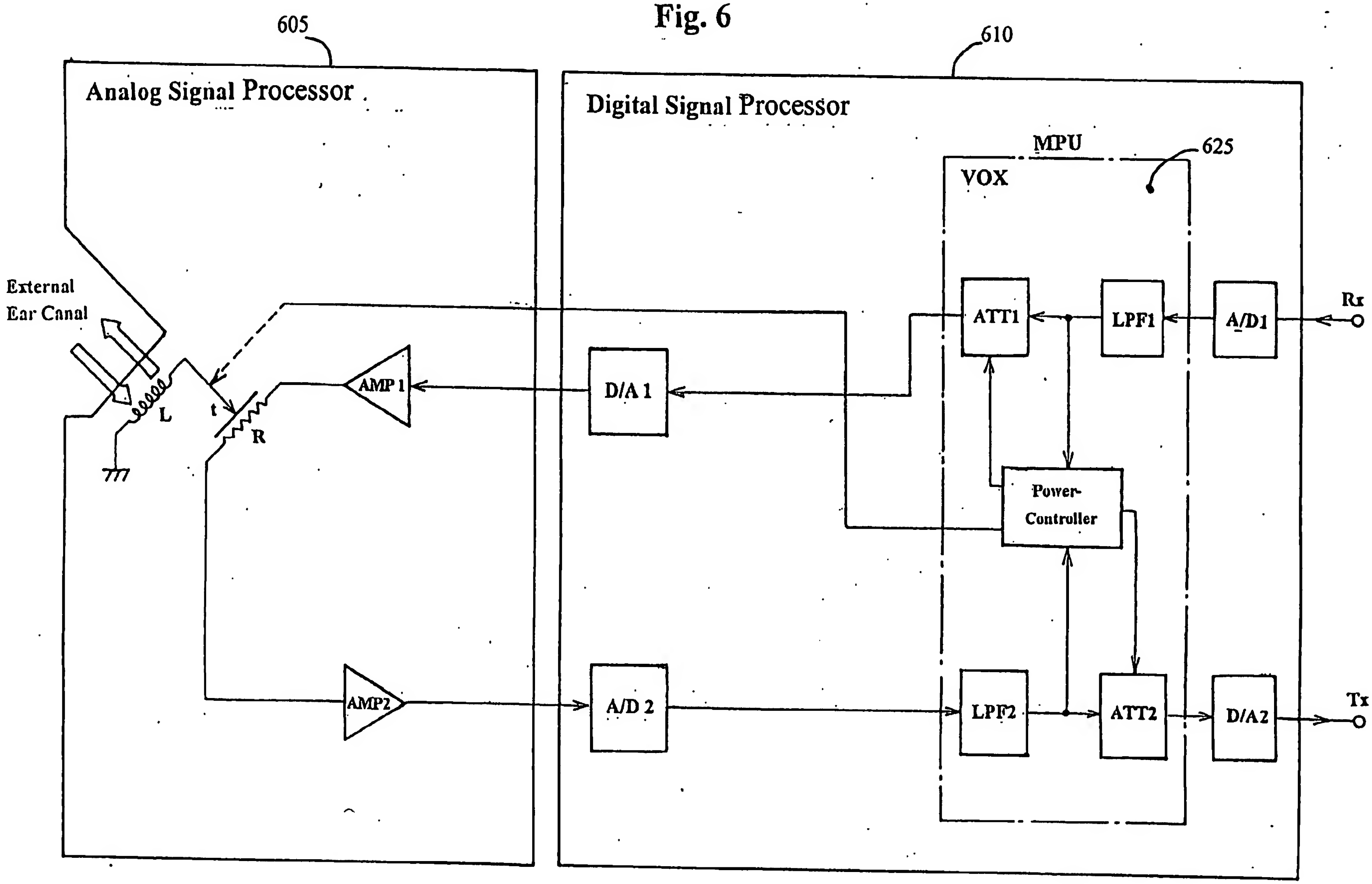

605
610
Analog Signal Processor
Digital Signal Processor
MPU
625
VOX
External
Ear Canal
L
t
R
AMP 1
AMP2
D/A 1
A/D 2
ATT1
LPF1
A/D1
Rx
Power-
Controller
LPF2
ATT2
D/A2
Tx

Fig. 7

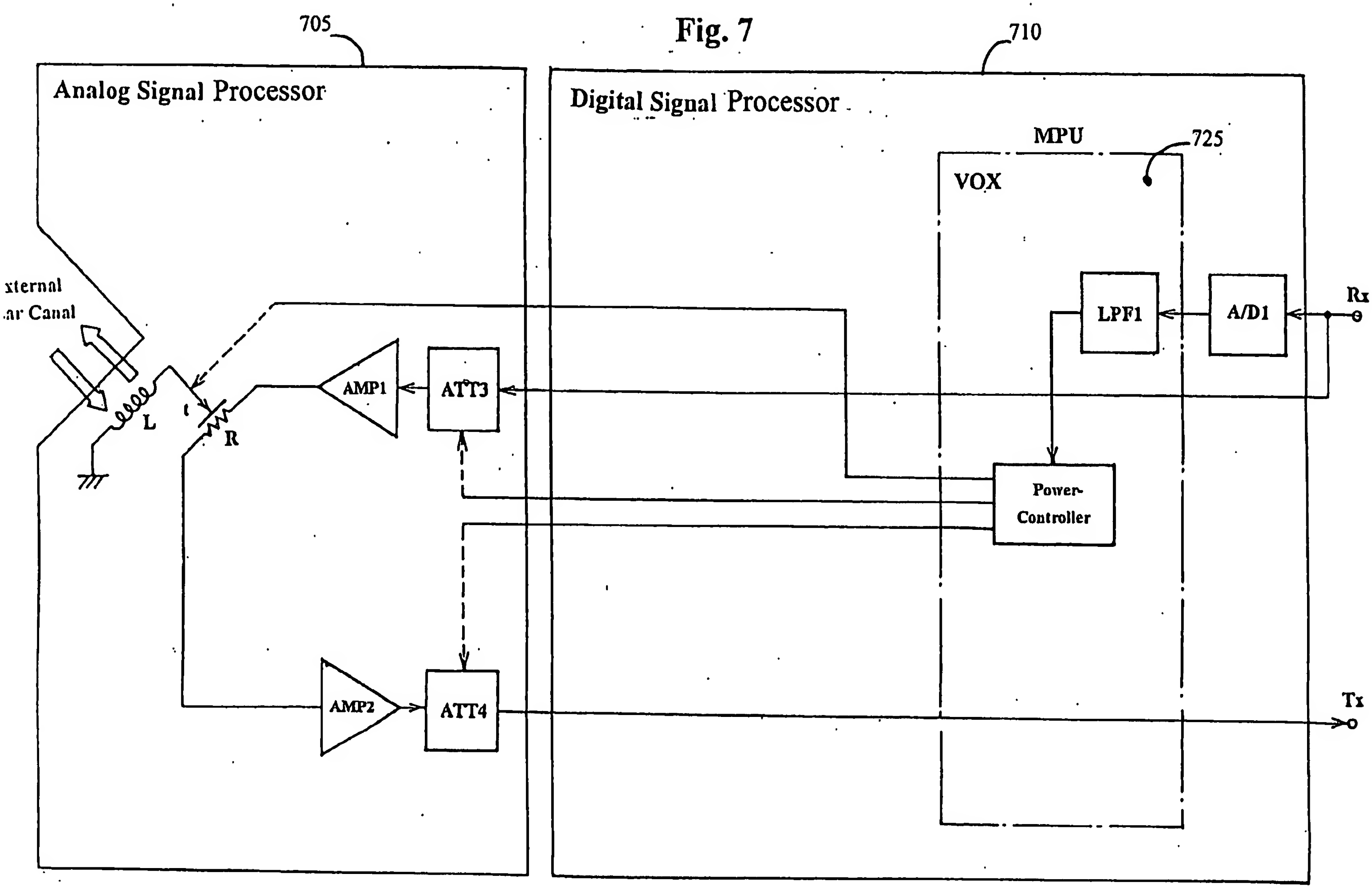

705
710
Analog Signal Processor
Digital Signal Processor
MPU
725
VOX
LPF1
A/D1
Rx
Power-
Controller
AMP1
ATT3
AMP2
ATT4
Tx
L
R
xternal
ar Canal

Fig. 8

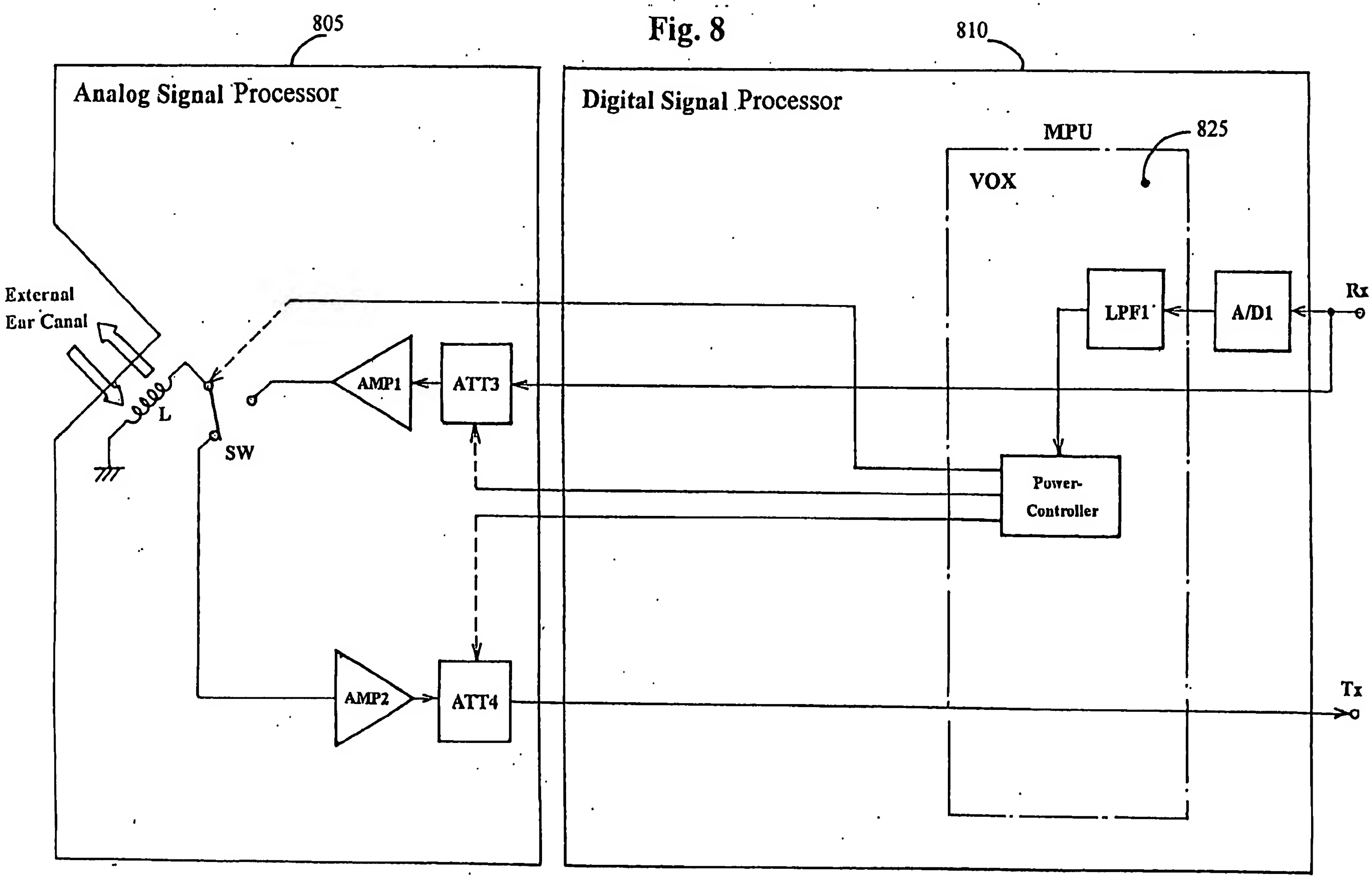

805
810
Analog Signal Processor
Digital Signal Processor
MPU
825
VOX
External
Ear Canal
L
SW
AMP1
ATT3
AMP2
ATT4
LPF1
A/D1
Power-
Controller
Rx
Tx

Fig. 9 (a)

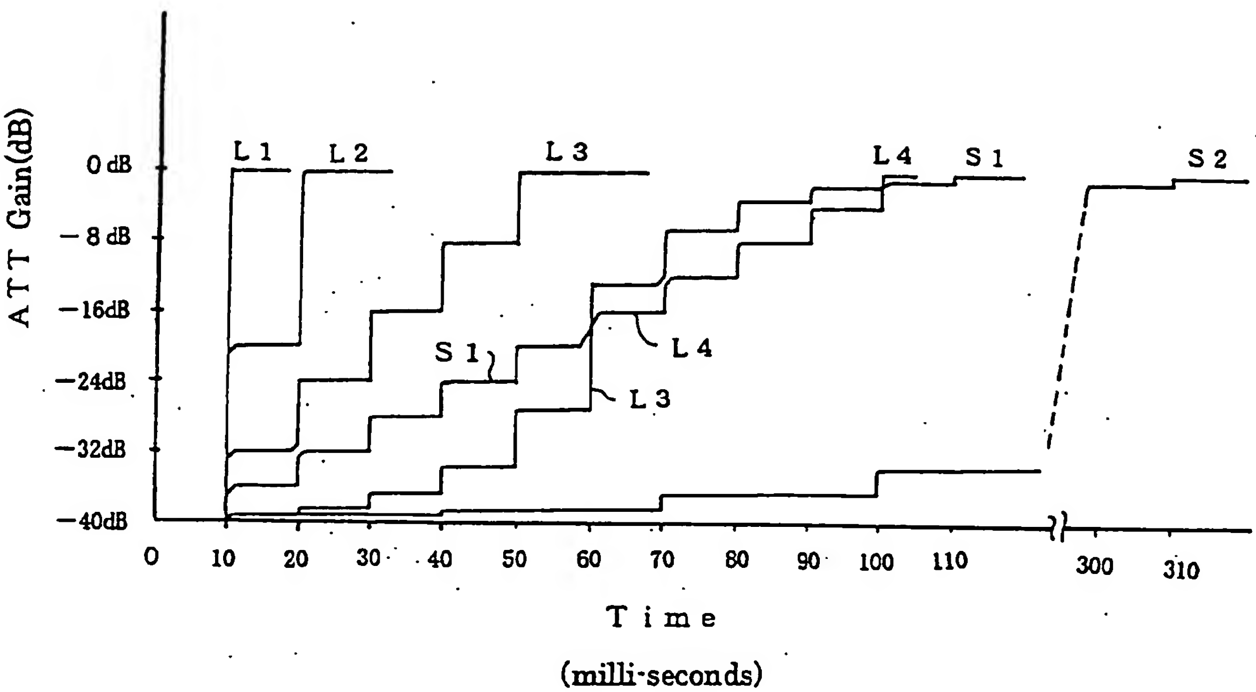


Fig. 9 (b)

Voice Power (dBm0)	L 1 (Δ=40dB)	L 2 (Δ=20dB)	L 3 (Δ=8dB)	L 4 (Δ=4dB)	S 1	S 2
30(loud)	C	C	C	C	C	C
24	C	C	C	C	C	C
18	C	C	C	C	B	B
15(regular)	C	C	C	C	A	B
12	C	C	C	C	B	B
6	C	C	C	C	C	C
0(whisper)	C	C	C	C	C	C

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